Building A PAiA 6740 "Hot Springs" Reverb

By Craig Anderton

WAIT! Don't turn the page. I know that simple spring reverb systems may not have the greatest reputation in the world, but this version uses a truly novel design technique. The result is a reverb system that offers an attractive combination of low cost and high performance.

You don't have to take my word for it, though. An engineer for a well-known manufacturer of effects boxes recently developed an all-electronic reverb system; part of his market research involved checking out the reverb market to see how his design compared to other currently available models. Since I felt he could be a little more objective about the "Hot Springs" reverb than I could be, I asked him to give it a listen. He was absolutely floored, and said it sounded better than anything else he had heard during his months of testing! I think you'll probably feel the same way after hearing it... but before we get into building, we need to examine just why the "Hot Springs" reverb (or "HS" reverb for short) is so different from the norm.

How Spring Reverbs Work

Let's begin by refreshing our memory as to how spring reverbs work in general *(see Figure 1)*. The spring connects to two transducers, one at the input and one at the output. Signals appearing at the output of the drive amp couple into the input transducer, which then takes this signal and couples it to a long spring; the signal is delayed as it travels down the spring. The output transducer picks up this delayed sound, and feeds it to a recovery amp which takes the extremely weak output of the reverb spring and amplifies it to a useable listening level.

So far, what we have described would only give a single "slapback" type of echo if it weren't for one very important fact: once the signal has reached the end of the spring, it bounces back along the spring towards the input, then reverses direction and bounces back towards the output again (contributing another echo), returns again towards the input, and so on until it eventually fades out. This creates the effect of multiple echoes and reflected sounds—just like you get in a large room. Also, there are several mechanical resonances in the spring itself that add peaks and dips in the response. This helps to simulate even more closely the properties of "real-world" reverb.

However, there are some problems (aren't there always!). The first is that

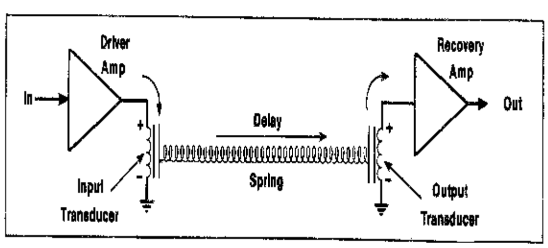


Figure 1

the motion of the spring itself adds a certain type of sound to the audio signal, which produces the characteristic "boing" and "twang" of spring reverbs. The second problem is that if you just listen to the reverb output, you'll hear a mushy version of the "dry" sound along with the sounds created by the multiple reflections and echoes we mentioned earlier. The third problem is that the spring output is in the millivolt range, which is exceedingly weak. As a result, the recovery amp must run at a very high gain to bring this signal up to a useable level, and this contributes noise to the system. The final problem we'll discuss is that springs have an inherent bandwidth limitation, which means that there is no significant audio energy above approximately 5 kHz. This is why springs often sound bassy and boomy compared to a good, crisp plate system.

Solving these Problems

The HS design uses "hot rod" guitar pickup technology to overcome the above-mentioned problems. This is one of those situations where the solution seems so obvious you wonder why no one has thought of it before; but to the best of my knowledge, and several other people, the following represents a totally original approach . . . so *MR&M* readers, you heard it first.

The basic principle is to take two springs and connect the input and output transducers in a special way, as shown in *Figure 2.* The input transducers are connected in series and *out-of-phase;* the output transducers are connected in series but *in-phase.* As a result of the out-of-phase input connection, the original audio signal—as well as the "sproings" and "boings"—cancel each other out at the output, leaving mostly the multiple echoes and reflections. This neatly solves problems 1 and 2, and gives a very rich reverb sound. Additionally, the input transducers are driven by a constant-current source that provides equal drive for high and low frequencies. This gives the bright high end associated with plate systems, while de-emphasizing the muddy, bassy sound often encountered with some spring reverb designs. Finally, by connecting the output coils in phase and in series, we double the overall output level. This means that the recovery amp doesn't have to provide quite as much gain, thereby giving an improved signal-to-noise ratio.

You might wonder why the cancellation effect discussed above doesn't cancel the *entire* reverb signal. Luckily, although reverb springs are matched closely enough so that the "boings" and dry signal are mostly cancelled, there are enough differences in response (particularly in the high frequency regions) so that the subtler reverb sounds are left pretty much unaffected.

Musically speaking, we traditionally think of reverb as trying to simulate the sound someone sitting in the audience would hear at a concert. However, anyone who has played on the stage of a 2000 to 10,000 seat venue knows that reverb sounds quite different from the performer's perspective; it is this sound which the HS reverb simulates. Instead of hearing an ill-defined reverb mix of dry signal and hall acoustics (as you do in the audience), from center stage you hear the reverb coming back at you without any discernible dry signal. What this means in the studio is that the HS reverb sound never "steps on" the signal being reverberated, since it contains the multiple reflections and echoes associated with a good reverb sound *while excluding virtually any trace of the original signal.* This is highly desirable, since in practice the reverb signal is mixed in at a low level compared to the signal being reverberated. By cancelling the muddy sounding version of the dry signal that comes out of most spring reverbs, the overall sound is clean, crisp and well-defined instead of being boomy and sproingy. Vague terms, to be sure, but if you've worked with inexpensive spring reverb systems in the past I'm sure that all the above expressions will sound familiar.

Controls and Options

Actually, the only control for the reverb unit is a level-matching trim pot. At maximum sensitivity, signals greater than −10 dB will overload the driver amp. At minimum sensitivity, clipping does not occur until the input reaches +15 dB or greater. There is an additional clipping indicator LED that lets you know when the driver amp is being overloaded. Due to the high-frequency boosting action of this stage, clipping will occur sooner at higher frequencies than at lower frequencies.

Finding Parts

As mentioned in the last D.I.Y. ("do-it-yourself") Limiter article *[Modern Recording & Music, November 1979],* whenever possible I try to line up a parts source that stocks all of the parts necessary to build a given circuit, as well as provide a repair service for wiring jobs that go astray. For this project, PAIA Electronics (1020 W. Wilshire Blvd., Oklahoma City, OK. 73116) is again providing this service—particularly because they stock the Accutronics model #1FB2B1D reverb springs which are used in this project. These springs were chosen for their low cost, small physical size, sound quality and ready availability from PAIA. As a result, all circuit components were selected with these springs in mind. While other springs may be used with this project, I cannot guarantee the quality of performance if substitutions are made. All other parts are commonly available and shouldn't be hard to find at all. If you decide to build the

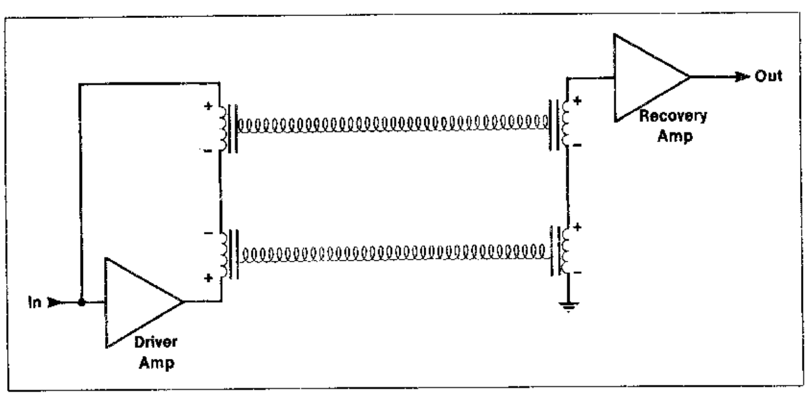


Figure 2

project from scratch, however, I highly recommend that you follow the circuit board layout as closely as possible to prevent ground loops, hums, oscillations and other potential problems.

Preliminaries

While this is a fairly simple project, some aspects of it (such as modifying the reverb springs) require a bit of skill. So, I wouldn't recommend that beginners undertake this project unless they have successfully completed similar projects in the past. On the other hand, I said the same thing about the D.I.Y. Limiter and several beginners built it with no trouble at all. You are probably the best judge of your capabilities. However, there are some basics which *must* be observed, namely:

■ Use a low wattage (40 watts or less) soldering iron. *Do not use soldering guns!*

■ Use only rosin core solder designed for electronic work. Any kind of acid core solder, or use of flux, will ruin an otherwise good circuit board and some of the parts as well. Use the solder sparingly; don't blob it all over a connection, since that can cause shorts between adjacent circuit board traces.

■ The amount of heat used in soldering is very important. Too little heat can cause "cold" joints, where the solder's rosin is not sufficiently melted; this causes a high-resistance connection. On the other hand, too much heat will damage parts. I'd suggest holding the iron tip against the

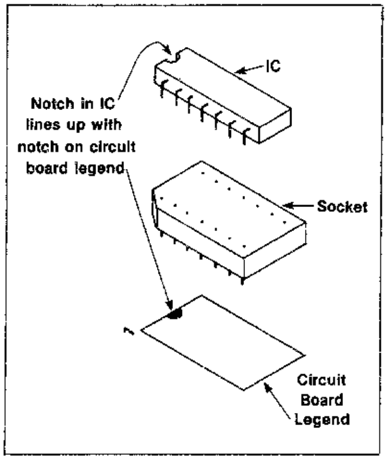


Figure 3a

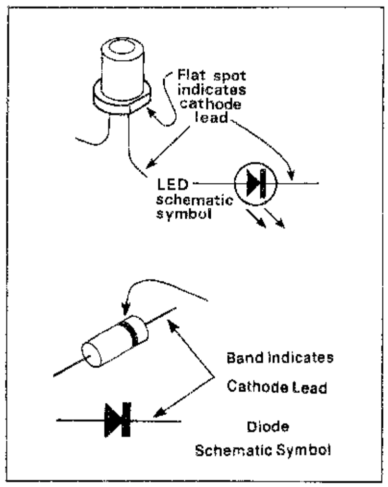


Figure 3b

connection to be soldered for a few seconds, then feeding in a little bit of solder and continuing to apply heat until the solder flows freely over the connection. If the solder balls up around the connection, reheat it and feed in a little more solder.

■ Use an IC socket. This simplifies replacement should the IC ever fail; also, you don't have to worry about frying the part through incorrect soldering techniques *(see Figure 3a)*.

■ Clean the copper side of the circuit board with steel wool to remove oxidation. A bright and shiny board contributes to successful soldering.

■ Note that electrolytic and tantalum capacitors are "polarized" components and have (+) and (−) marks, just like a battery. Like a battery, if you don't hook these parts up right the circuit won't function; so, the circuit board legend has a (+) symbol near the hold where the capacitor's (+) lead must go. LEDs and diodes are also polarized. Referring to *Figure 3b,* the LED symbol is an arrow pointing towards a bar. Generally, the bar (or cathode) end of the LED is designated with either a flat indentation in the case or a dot of paint. Diodes have a similar schematic symbol, and the bar end of the diode corresponds to a band painted on the diode itself.

■ Take your time and work carefully. Impatience is one of the biggest reasons why do-it-yourself projects fail.

■ No power supply is shown in the schematic. If you built the Limiter,

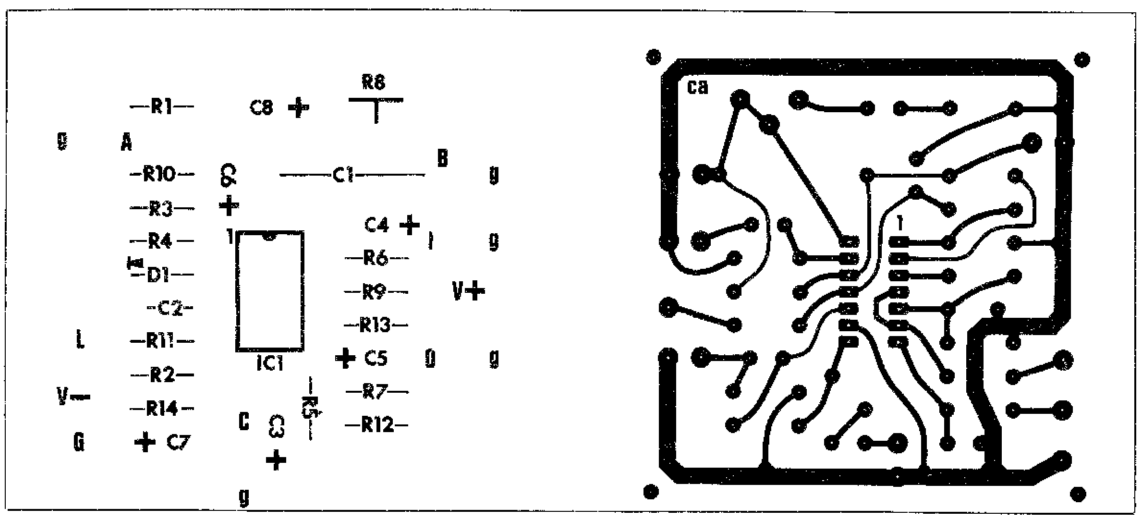


Figure 4

you can use the same power supply; simply tap off another set of connections for the reverb unit and you're ready to go. Otherwise, you can use any +/−15 V bipolar supply such as the PAIA 9770R or the HK-116 from Bill Godbout Electronics (P.O. Box 2355, Oakland Airport, CA. 94614).

Space prohibits us from going into all possible aspects of electronic construction. If you'd like to find out more about this topic, refer to my *Electronic Projects for Musicians* book (published by Music Sales, 33 West 60th Street, NY, NY 10023) *[Craig . . . is this a blatant plug for your book!?—Ed.]*. It contains complete information on finding parts, soldering, packaging projects, labelling, etc.

CONSTRUCTION: There are four distinct phases to construction: 1) loading and soldering the circuit board; 2) modifying the reverb springs; 3) packaging the springs and circuit board in a suitable chassis; and 4) connecting the circuit board to the power supply and springs. We will deal with each one in order.

SOLDERING the CIRCUIT BOARD: Referring to *Figure 4* (the component side of the board) and the parts list, solder the various components in place. Start with the resistors first, then the IC socket, capacitors and trimpot. Check that all solder connections are well made, then proceed to the next section.

MODIFYING the REVERB SPRINGS: In order to do the various in-phase and out-of-phase tricks mentioned in the beginning, we have to modify the wiring of the two reverb springs. This is probably the most complex part of the project, so pay careful attention to the following instructions.

Begin by placing the two reverb units side-by-side with the springs facing up, as shown in *Figure 5;* note that the two jacks are facing to the left. The input jacks are towards the top of this figure, and the output jacks towards the bottom. Since the input jacks are easiest to wire up, we'll do them first.

Referring to *Figure 6* (which shows the modified wiring), disconnect the black wire attached to each reverb spring's input transducer from the associated ground lug of the input jack. Next, note that there are some little springs that hold the spring plate to the case, and that these springs hook on to a hole in the side of the reverb spring case. Now connect a small piece of thin gauge insulated wire to the black lead of the left-most reverb unit,

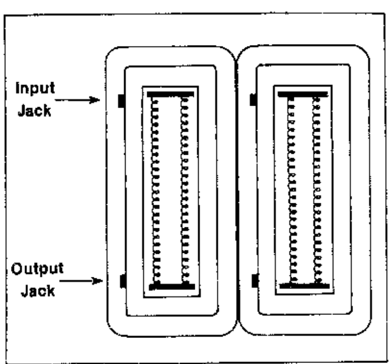


Figure 5

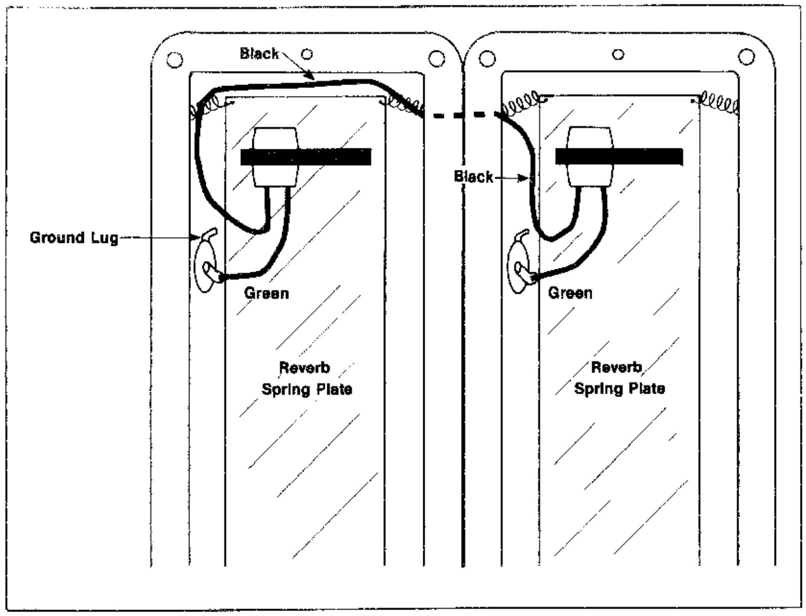


Figure 6

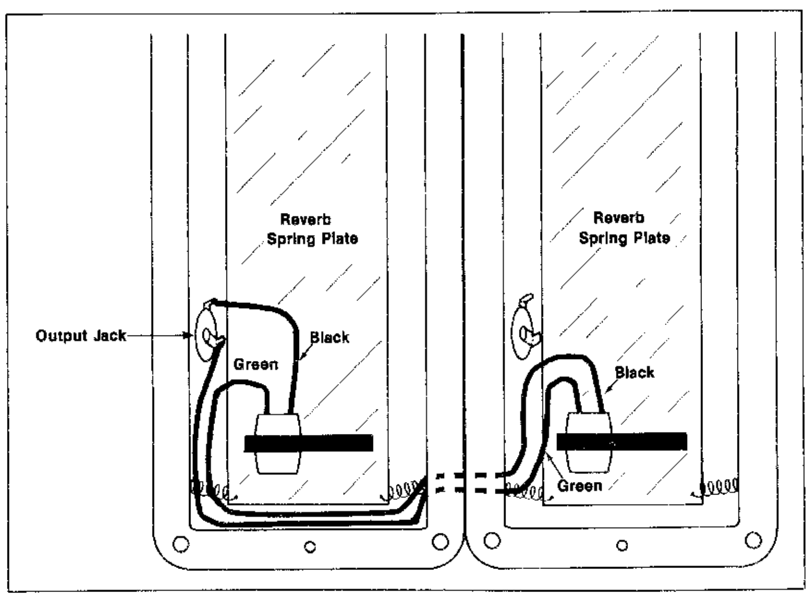


Figure 7

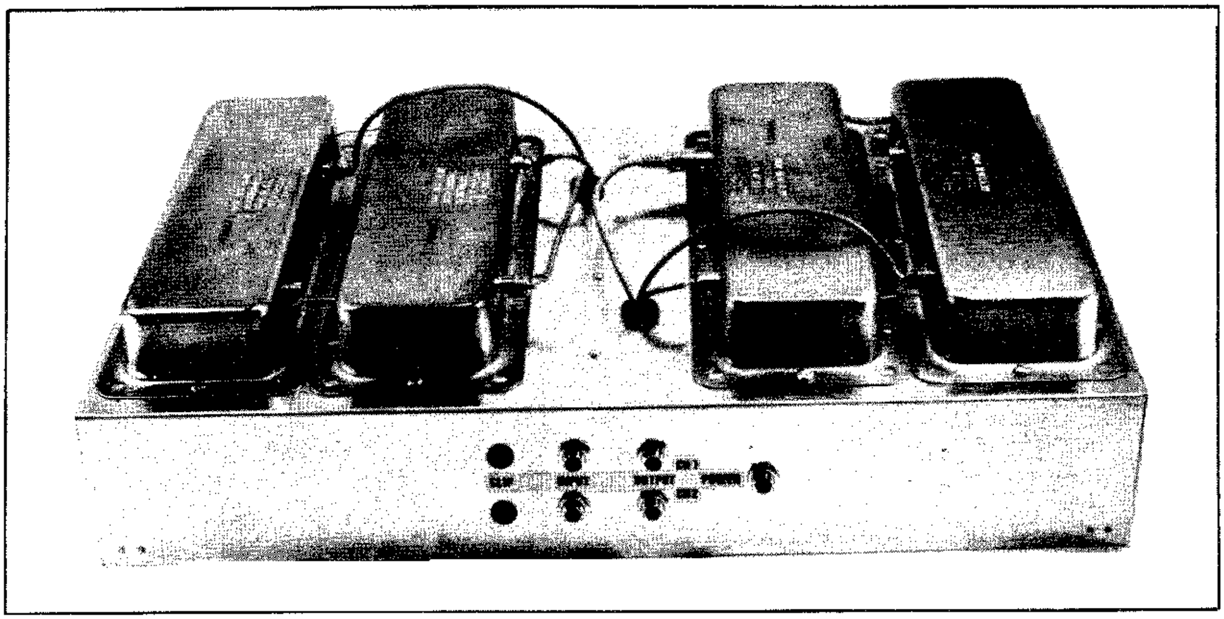


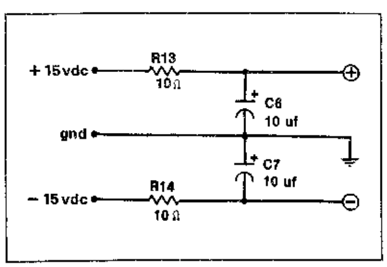
Figure 8

run it around the inside of the case as shown, and run it through the holes in both cases where the little springs hook in. Then, after you've gotten this wire inside the reverb unit on the right, connect it to the black wire coming from the remaining transducer. Finally, use a thin piece of electrician's tape to insulate the connection between the transducer leads and the added "jumper" lead. Check with *Figure 6* again to make sure everything is connected correctly, and that the extra length of wire does not interfere with the free motion of the reverb spring plate.

Figure 7 shows a detail of the modified output jack wiring. In this case, disconnect the green wire from the left transducer and the black *and* green wires from the right transducer from their associated phono jacks. Connect the green wire from the left transducer to a short length of thin gauge insulated wire, and run this into the right reverb case through the case holes (like the ones mentioned above). This wire should then connect to the black wire from the right transducer. Finally, the green wire from the right transducer should connect to a wire that again runs through the two holes used for routing the last wire, and ends up connecting to the "hot" terminal of the output jack mounted on the left reverb unit. Look carefully at *Figure 7* to check that all is well. It is important that the wires not interfere with the motion of the springs or the plate to which they connect. Make a mark on the case near the output jack so that you don't forget which one is wired to the springs.

(If you feel ambitious, the two wires connected to the output transducers can be shielded. However, you'll have a hard time finding shielded wire that's skinny enough to be comfortably routed as shown in the diagram—luckily shielding isn't absolutely necessary.)

Now that the springs have been modified, it's time to find a suitable enclosure to hold the springs and circuit board. *Figure 8* is a photo of the case I used, which is a general purpose aluminum chassis sold by electronics supply houses. Since I wanted stereo reverb, I used four springs (two for each channel). These springs should be mounted as shown; do not mount them upside down or sideways, as they don't sound right that way. In my particular case, I mounted the two circuit boards for the two channels inside the box, and ran the connections from the boards to the springs through a few holes drilled in the chassis. The input/output jacks and LEDs mount on the front of the box as shown.



Power supply connection

Connecting It All Together

Now we come to the last stage. Run a shielded cable from pad I on the board to the input jack; connect the shield at the board end only. Note that there is a pad next to pad I (pad "g") where you can connect the shield. Next, run a shielded cable from pad O to the output jack; again, connect the shield at the board end to the pad "g" next to pad O. All future steps involving shielded cable should have the shields connect to the nearest pad "g" on the board. Do not confuse these with point "G," whose use will be covered later.

Now connect a length of shielded cable to point A, and terminate it in an RCA phono jack. This wire should be long enough to reach either reverb spring input jack. The shield should *not* connect to the plug's ground, but just to pad "g" on the board. Plug this ungrounded phono jack into the reverb spring input. Then, in a similar fashion, connect a piece of shielded cable to point B, with its shield con-

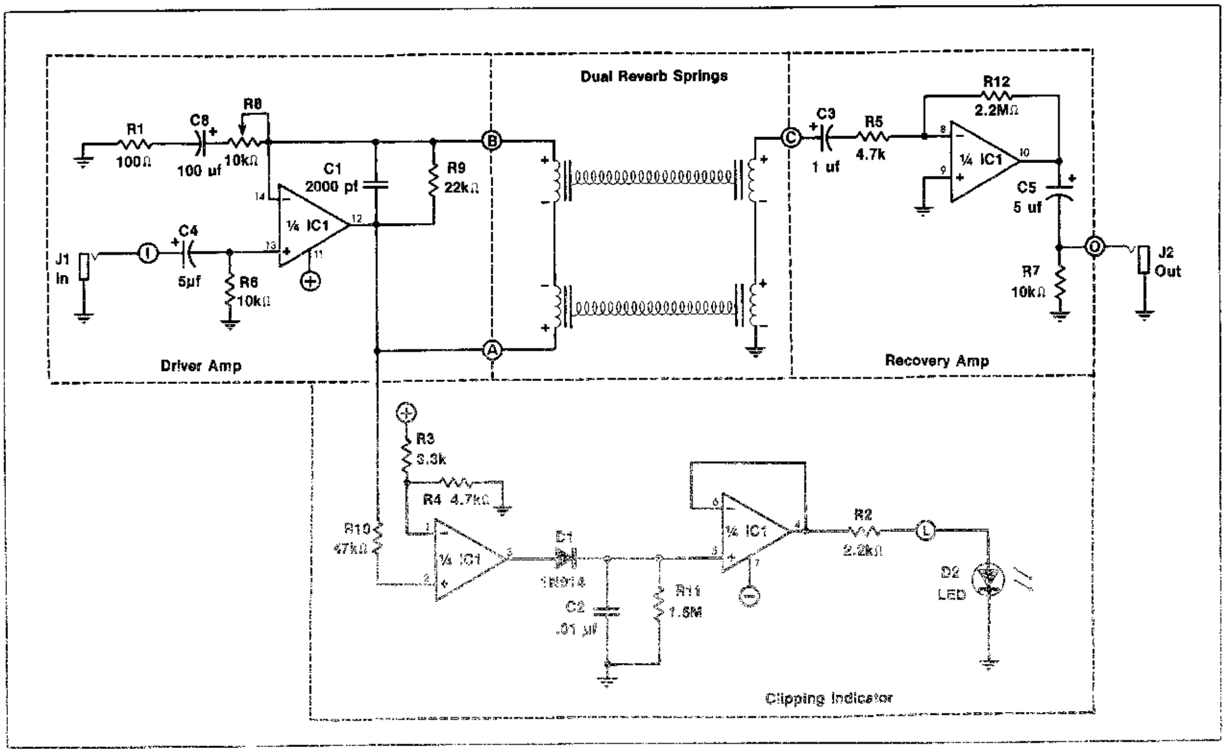


Figure 9

nected to the "g" pad near B. Again, check that the shield does *not* connect to the plug's ground. The ungrounded phono plug connected to this wire should plug into the remaining reverb spring input.

Our final piece of shielded cable connects to point C, with (you guessed it) the ground connecting to the nearest point "g." This wire should terminate in an RCA phono plug; *but this time, make sure that you do connect the shield to the plug's ground,* then plug into the reverb spring output jack you marked in an earlier step.

O.K., now we've connected the springs to the circuit board and the circuit board to the input and output jacks. Our final task is to hookup the LED and power connections. Run a wire from pad L to the anode of the indicator LED; run a wire from the cathode of the LED to a convenient ground point, such as the ground tab of the output jack. For power, connect pad G to the ground tab of the input jack, then connect the +15 V line from your power supply to pad V+, the −15 V line from your supply to pad V− and the ground line from your supply to the chassis ground or input jack ground tab. In my version, I used a stereo jack

SPECIFICATIONS

Maximum input before clipping: −10 dB (maximum sensitivity), +15 dB (minimum sensitivity)

Input impedance: 10 k (may be changed to 100 k by replacing R6 with a 100 k resistor and C4 with a .22 uF capacitor)

Output impedance: Less than 1 k

Current consumption: +7 mA, −7 mA

Signal-to-noise ratio (peak output compared to residual noise): greater than 63 dB

Frequency response of reverb signal: (please note, due to the various resonances and uneven response desirable in a reverb unit, it is difficult to give accurate response figures. The *Figure 10* graph is an attempt to average the response to give a meaningful composite figure).

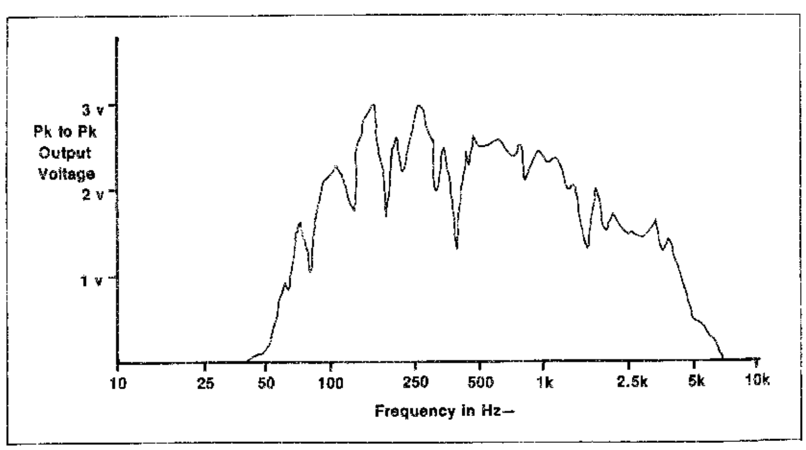


Figure 10

"HOT SPRINGS" REVERB PARTS LIST

Resistors

All resistors are ¼-watt, 10% tolerance unless noted. 5% tolerance resistors are preferred.

R1	100 Ohms
R2	2.2 k (2 k 2 metric)
R3	3.3 k (3 k 3 metric)
R4, R5	4.7 k (4 k 7 metric)
R6, R7	10 k
R8	10 k trim pot
R9	22 k
R10	47 k
R11	1.5 M (1 M 5 metric)
R12	2.2 M (2 M 2 metric)
R13, R14	10 Ohms

Capacitors

All capacitors are rated at 15 or more working volts.

C1	2000 pF or 2200 pF (2 nF metric) polystyrene
C2	.01 uF (10 nF metric) disc ceramic
C3	1 uF electrolytic or tantalum
C4, C5	5 uF electrolytic or tantalum
C6, C7	5 to 50 uF electrolytic or tantalum
C8	100 uF electrolytic or tantalum

Semiconductors

D1	1N914 or equivalent silicon diode
D2	Red LED
IC1	RC4136 (Raytheon) or XR4136 (Exar) quad op-amp

Other Parts

J1, J2	Open circuit ¼-inch phone jacks or RCA phono jacks (depends on your particular setup)
Misc.	14 pin, IC socket, Accutronics #1FB2B1D reverb spring, circuit board, solder, case, wire, etc.

on the front for my power supply wiring; this enables me to use a stereo cord to plug the reverb unit into a multiple-outlet power supply.

One more thing: If for some reason you mount the reverb springs on a non-conductive material (e.g., plastic), run a wire from each reverb spring case to ground. It is important that the spring cases be grounded to keep hum to a minimum.

Testing Time

Connect the output of the reverb system to a suitable monitor amp (with the volume turned down!), then patch an instrument, tape track, or similar signal source into the reverb input and apply power. Turn up the monitor; you should hear the reverberated sound. Now observe the indicator LED. If it doesn't glow very much, *increase* the sensitivity trim pot so that it flashes on signal peaks in order to avoid excessive noise. If, on the other hand, the LED flashes a lot, *decrease* the sensitivity to avoid distortion. The setting of this trim pot is rather important, so don't be afraid to experiment. If you find that the reverb output is too noisy, make sure that the sensitivity control is set properly in order to give the maximum possible level to the springs short of distortion.

How It Works

Referring to the schematic *(Figure 9)*, IC1A is the driver amp. Capacitor C1 tunes the reverb for a response peak at about 5 kHz, while R9 sets a ceiling on the maximum amount of gain generated by this stage. IC1C and IC1D tap off the output of this stage and comprise a simple clipping indicator. If the signal appearing at IC1's output exceeds the threshold set by R3/R4, then IC1C turns on and charges C2 through D1. C2 acts as a pulse stretcher to catch short duration transients, with the decay time being set by R11. IC1D simply buffers this cap and drives the clipping indicator LED.

Signals appearing at the spring output drive IC1B, the high-gain recovery amp stage. The 2.2 M feedback resistor is kind of extreme, but of all the configurations tested this one gave the lowest overall noise figure. R5 is a low enough value to load down the springs just a tiny bit, which reduces excessive high frequency response that would otherwise add a kind of "tinniness" to the sound. You can substitute a 10 k resistor for R5 if you'd like to trade off more noise for extended bandwidth, but I think 4.7 k gives the best overall results.

By the way, if the reverb unit has poor lead layout or ground loop problems (which it shouldn't if you followed the instructions carefully), it's possible that IC1B will oscillate. To avoid starting over from scratch, you can fix the problem by adding a 10 to 20 pF capacitor in parallel with R12. However, this should not be necessary if you grounded your shielded cables correctly and used the circuit board layout shown in *Figure 4*.

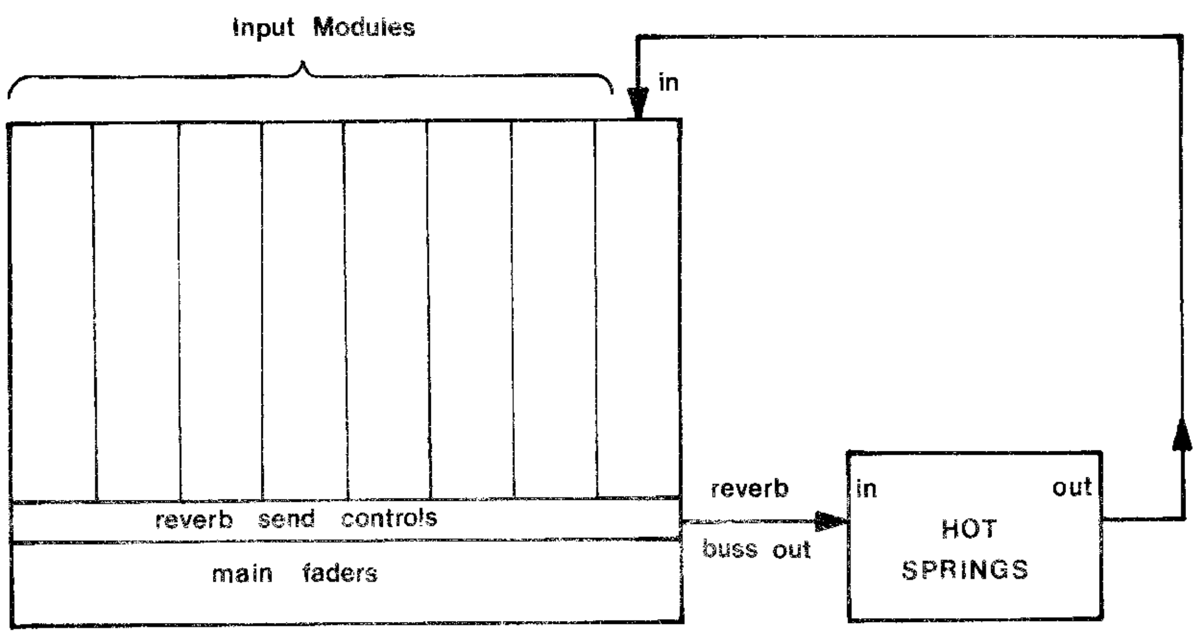
In Conclusion

I hope you get as excited about this reverb unit as I am; I think it sounds real good, and am happy to be able to share it with the recording fans who devour *MR&M* each month. If you have any questions about the reverb's operation, or run into difficulty, be sure to write so that we can cover any problems in future issues of *MR&M*.

Applying the Hot Springs: Recording console. Figure A-1 shows a typical mono Hot Springs application. The reverb buss output feeds the Hot Springs input; the output of the Hot Springs then proceeds to one of the mixer inputs. This input then becomes dedicated to reverb. Turning up the master fader associated with the reverb channel brings reverb into the mix. Generally, you'll want to pan this in the center of the stereo spread.

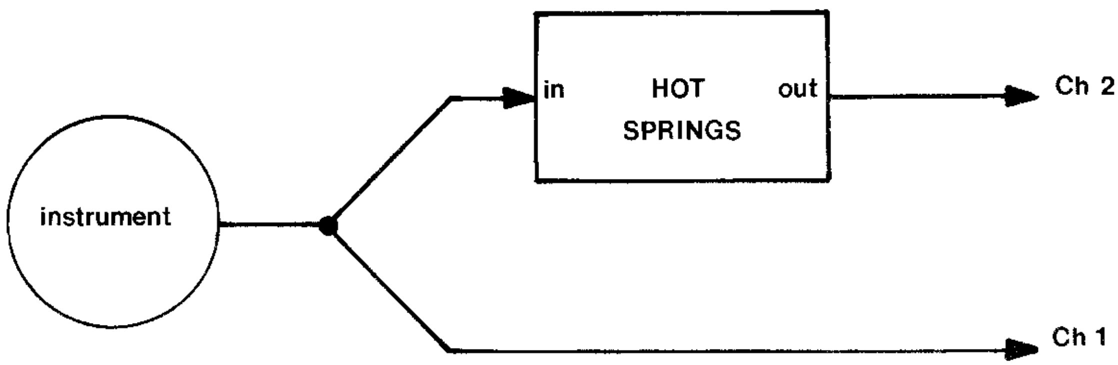
If you have a stereo Hot Springs setup, and your mixer has a stereo reverb buss, you can feed two reverb units and then dedicate two mixer inputs to stereo reverb returns. In this case, you'd probably want to pan the reverb channels to the left and right of center to achieve a good stereo spread.

Fig. A-1



Applying the Hot Springs: Instrument applications. Figure A-2 shows a typical setup with a two channel amp. Note that for instrument use the Hot Springs should be modified as mentioned in the instructions under "Specifications" for a 100k input impedance; for an even higher impedance (suitable for guitar), change R6 to 220k and C4 to 0.1 uF. Use a Y cord or splitter to send the instrument signal to channel one of your amp as well as the Hot Springs input. Then patch the Hot Springs output to channel two of the amp. Adjust the amount of reverb with channel two's volume control, and adjust equalization to suit.

Fig. A-2



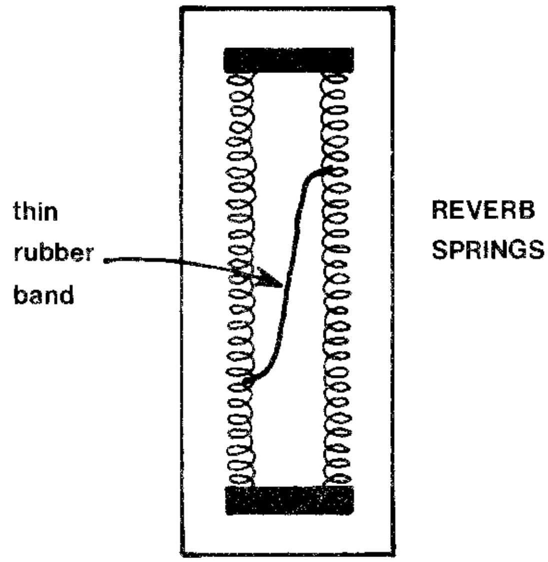
Hot Springs limitations. The Hot Springs has been successfully installed in hundreds of audio applications, and is recognized as a great sounding, cost-effective spring reverb. However, it is still a spring reverb at heart, and therefore may produce a small "glitch" when a sudden, short transient enters the unit. Generally, this is only a problem if you're listening to the reverb channel by itself because whatever causes the glitch will be loud enough to mask that glitch, as long as some of the dry signal is present along with the reverb sound. Luckily, the Hot Springs design minimizes this glitch, but it still can be noticed under worst-case conditions.

Another limitation of spring reverb is "flutter". This can occur with the Hot Springs if the unit is overloaded, driven by a loud transient, or driven by something with little or no harmonic content that covers a wide frequency range (such as a sweeping sine wave audio oscillator). If you avoid overloading the unit and put complex signals (voice, guitar, bass, program material) through the Hot Springs, this problem will be virtually inaudible.

Also note that the spring units will not sound as good if mounted upside-down or on their sides.

Modifying "room size". The Hot Springs sounds like a big hall or auditorium. If you want to simulate a smaller room, and minimize flutter as well, add some damping. Take a long, <u>thin</u> rubber band and wedge one end in between the spring coils about 1/3 to 1/2 way from the end of one spring, and wedge the other end in the other spring in an opposite manner (see figure A-3). You can also "tune" the springs to give a more metallic effect by wedging a 1/2" or so resistor clipping in between the spring coils. Varying the placement of this clipping alters the "tuning". Please note that the reverb transducers and springs are delicate, so be careful and don't void your warranty by damaging the springs.

Fig. A-3



Using the Hot Springs with special effects. There are several ways to enhance the Hot Springs sound:

- **Followed by EQ.** The PAIA Parametric Equalizer is ideal for post-processing of the Hot Springs and changing the reverb timbre.
- **Followed by flanger or chorus unit.** Flangers and chorus units, such as the PAIA Hyperflange + Chorus, add animation and interest to the sound of the reverb. Stereo chorus units are excellent for synthesizing a stereo reverb field from a single Hot Springs channel.
- **Preceeded by delay line.** Patching a delay line before the Hot Springs allows you to add pre-delay in the range of 25 to 100 ms (these are starting points; whatever delay works musically is fine). The PGS Electronics Chorus unit does a good job for this application.
- **Preceeded by limiter.** While the Hot Springs is capable of handling fairly large input signals, with excessively large signals distortion can result. Patching a limiter in front of the reverb (such as the PAIA Dual Limiter), and clamping the input signal just below the point of distortion, prevents overload problems.

* * * * *

Thank you for choosing PAIA equipment. The Hot Springs has proven to be a very popular kit, and we hope that you enjoy using it in your audio setup.